REMARKS

By this Amendment the specification has been amended to better conform with U.S. practice, claims 3, 5, 6 and 7 have been amended to depend from only one preceding claim, and claims 2, 4 and 8 have been reworked. Entry is requested.

In the outstanding Office Action the examiner has rejected claims 1-3 and 8-11 under 35 U.S.C. 102(b) as being anticipated by Kalin et al.[1] The applicants disagree.

Kalin et al. disclose use of an adaptive compensation filter to compensate for acoustical mechanical disturbance feedback between an electrical-acoustical converter and an acoustical-electrical converter of a hearing aid.

The examiner asserts that Kalin et al. disclose a method for cancelling feedback in an acoustic system comprising, *inter alia,* (a) using a high pass filter to prevent low-frequency signals from entering the LMS algorithm; and (b) using an additional feedback cancellation filter and a noise generator for providing low-frequency input for the LMS algorithm.

A high pass filter (Fig. 13, 108) is disclosed in Kalin et al., but as explained in detail in column 11 the effect of preventing low frequency signals from entering the LMS algorithm is not achieved by this filter. The

[1] The examiner has incorrectly evaluated claims 1-13 as found in WO 01/06812A1 instead of the substitute claims 1-9 attached to the IPER of 24 October 2001.

filter 108 is a part of a modelling unit which models the behavior of the loudspeaker 9. As explained in column 11, lines 19-57:

"The modelling unit comprises a prefilter 100 with a transfer characteristic F_1 (Ω) being substantially a low path characteristic. The corner frequency Ω_1 of the Bode diagram schematically shown in prefilter block 100 is approximately 0.8 kHz in a preferred embodiment, the gain $|F_1|$ at this corner frequency Ω_1 approximately 0 dB. The slope S_1 is approximately 0 dB/DK.

"The identification entities, namely corner frequency Ω_1 and the slopes S_1 and S_2 as well as the gain, e.g., at the corner frequency Ω_1, are found by identification of the loudspeaker or EAC 9 to be modelled.

"Following the prefilter 100, there is provided a linear amplification unit 102 at which the amplification factor K is set. Following the linear amplification unit 102, there is provided a not-linear amplification unit 104. The transfer characteristic of the not-linear amplification function Y=Q(x) is e.g.:

$$y=x+ax^2+bx^3+cx^4+dx^5.$$

"For small input signals, the amplification of the not-linear amplifica-tion unit 104 is unity, so that the amplification characteristic adjacent to the origin has the slope 1. For larger input signals x the not-linear amplification characteristic has, as is known from loudspeakers or from EAC 9, saturation characteristic.

"The coefficients a, b, c, d of the not-linear amplification characteristic and the amplification factor K are determined by identifying the converter to be modelled.

"Following the not-linear amplification unit 104, there is provided a linear amplification unit 106, whereat the amplification K of the linear amplification unit 102 is compensated, K^{-1}. Following the unit 106, there is provided a filter unit 108 substantially with high pass characteristic, which, as is shown in FIG. 13, substantially compensates the frequency characteristic of the prefilter 100.

"Thus, the converter modelling unit, i.e., the loudspeaker or EAC 9 modelling unit as shown in FIG. 13, comprises substantially a linear amplification part formed by the units 100, 102, 106, 108 and a not-linear amplifier unit 104."

From this disclosure, it is clear that the filter 108 is inserted in order to compensate for the effect of filter 100 provided earlier in the signal path. Thus, the overall function of the two filters 100 and 108 will not prevent low-frequency signals from entering the LMS algorithm. The amplitude of filter 108 at frequencies below Ω_1 is approximately 0 db and thus in this frequency range no attention of the signal will result from the action of the filter 108. Although the filter 108 is described as having a high pass characteristic, this filter will not prevent low-frequency parts of the signal from entering the LMS algorithm and thus it is new over Kalin

et al. to have a highpass filter to prevent low-frequency signals from entering the LMS algorithm.

According to the examiner, Kalin et al. further discloses "an additional feedback cancellation filter (Fig. 15, No. 5f) and a noise generator (Fig. 15, No. 127) is used for providing low-frequency input for the LMS algorithm."

The applicants do not agree that the Kalin et al. disclose a method with such elements. Firstly, the feedback cancellation filter 5f is not "additional." Secondly, the filter 5f in Fig. 15 is an amplification filter unit, and not a feedback cancellation filter. The feedback cancellation filter 15f in Kalin et al. is arranged to receive the noise signal injected at summation point 129, and further this noise signal is also fed directly into the speaker unit 9. As explained in col. 13, lines 34-40, this noise may be audible and special means has to be provided to keep the noise level below the audible level.

According to the present invention an additional feedback cancellation filter is arranged along with the noise generator and this arrangement is used to provide the required low-frequency input to the LMS algorithm. The use of the additional feedback cancellation filter allows the noise signal to be used without feeding this signal into the sound path of the hearing aid. The noise signal thus does not become audible and special means to avoid audibility of the noise signal are not needed.

Kalin et al. do not disclose the use of a high pass filter to prevent low-frequency signal from entering the LMS algorithm, and also it is not disclosed to provide an additional feedback cancellation filter and a noise generator used for providing low-frequency input for the LMS algorithm. The advantages achieved by these novel features are described in the specification at page 8, lines 19-23, where the advantage of the high pass filter is to attenuate frequency ranges in which the autocorrelation of the external input signal is long and thereby reduce the possibility of poorly adjusted coefficients and poor sound quality, and further on page 9, lines 4-27, it is explained how the additional feedback cancellation filter and the noise signal allows the LMS algorithm to remain stable. None of this is derivable from Kalin et al. in any obvious way.

With regards to claim 2, no sign swapping algorithm is disclosed in Kalin et al.

Also, the use of a steep low pass filter as mentioned in claim 3 is not disclosed in Kalin et al.

The examiner's rejection based on Kalin et al. should be withdrawn.

The examiner has rejected claims 12 and 13 under 35 U.S.C. 103(a) as being unpatentable over Kalin et al. in view of Yoshida.

However, no claims 12 or 13 are presented for evaluation.

Favorable reconsideration of this application is requested.

Respectfully submitted,

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